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UNITED TECHNOLOGIES CORPORATION

Stratford, Connecticut 06601-1381

Contract NAS1-17146
Task Assignment 7

April 1988

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SIGNAL RESTORATION ON NON-STATIONARY ACOUSTIC SIGNALS IN THE TIME DOMAIN

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1.0 SUMMARY

The signal restoration technique herein described is a method of transforming a non-stationary signal acquired by a ground based microphone to an equivalent stationary signal as if the microphone were moving with the rotorcraft. The benefit of the signal restoration, when used in conjunction with a linear microphone array, is a simplification of the flight test requirements because, in many cases, it could dispense with the need to acquire acoustic data with another aircraft, such as NASA's YO-3, flying in concert with the rotorcraft. Using the ground based microphones, the aircraft can also be tested over its full speed range. The data quality is also generally improved because the ambient noise (the contamination of the signal by the propeller and wind noise, hence the signal to noise ratio) is much better. The restoration methodology presented in this report can also be combined with other data acquisition methods, such as a multiple linear microphone array averaging technique, for further improvement of the final results.

This report contains the methodology and software description for performing the signal restoration in the time domain. The method has no restrictions for flight path geometry or flight regimes; it requires only that the aircraft spatial position be known relative to the microphone location and time coded for later synchronization with the microphone data. However, for best results the signal propagation path length should be limited to 1 km and ground plane microphones should be used because atmospheric and ground plane effects are not included in the restoration process at this time.

The restoration process assumes that the moving source radiates a stationary signal, which is then transformed into a non-stationary signal by various modulation processes. The frequency and amplitude modulation mechanisms are treated according to their effect on the signal. The signal modulation mechanisms, (source and medium motion, the heterogeneity of the atmosphere and the ground effects) are enumerated and their respective effects on the signal propagation are assessed.

The restoration procedure, at this stage of development, contains only the modulation mechanisms which are due to the source motion. The other modulation mechanisms, the non-stationarity of the

propagation medium and ground effects are not included in the present version of the restoration procedure, however an assessment of the effects on accuracy by the mechanisms which are not included in the restoration process has been made. A list of the mechanisms which should be added to the restoration process has been compiled. This list is based on the relative importance of a modulation effect, the incremental benefit of its inclusion, and a possible implementation methodology. In addition, the report contains a short review of the past and current efforts in the field of signal restoration and an assessment of the performance and ease of implementation of various restoration methodologies.

1.1 Results and Recommendations

Results:

- A signal restoration technique, which allows the transformation of a non-stationary acoustic signal acquired by ground based microphones to an equivalent stationary signal, has been developed and applied to non-stationary microphone data acquired from a Sikorsky S-76A flyover. The capability of this technique to discern signal details, which would not have been otherwise identifiable, has been demonstrated.

- The restoration technique has been shown to work well when the results are viewed in the frequency domain. However, attempts to average the signal in the time domain were mostly unsuccessful and this problem is attributed to the lack of high resolution main rotor speed data. The main rotor speed correction was based on sampling of the rotor speed every 0.5 seconds. It is apparent that this is insufficient because the rotor has a cylic speed variation even during one revolution. To overcome this problem a shaft encoder should be added so that the rotor speed could be recorded. Then a signal proportional to the rotor speed would be available for restoring the fine temporal structure of the time domain signal.

- In addition to the sharper, more well defined, main and tail rotor tones, well defined interaction tones are extracted from stationary microphone flyover data by using this restoration method.

- The program for performing the restoration process has been written in a modular form to facilitate the program installation on various computers.

- There remains a residual smear in the spectrum, which is probably due to the modulation mechanisms which were not included in the restoration process.

Recommendations:

- The restoration technique should be applied to the microphone array data to investigate if further improvements can be achieved.

- The rotor speed should be measured using an encoder. It is recommended to use an encoder which would produce 1024 or 4096 pulses per revolution.

- Considerations should be given to the inclusion of other modulation mechanisms and ground effects. The proposed improvements are listed in order of their perceived importance:

    (a) Modeling the source as a distributed source and for distances less than 100m.

    (b) Atmospheric Absorption.

    (c) Small scale turbulence.

    (d) Refraction effects.

    (e) Ground plane effects.

- Correlation between the acoustic data gathered by using the in-flight measurement technique and the restored ground based microphone array data should be conducted.

2.0 INTRODUCTION

The rotorcraft acoustics engineer needs reliable flight data to identify the sources of the noise generated in different flight regimes to formulate appropriate noise reduction measures. Furthermore, to develop accurate noise prediction methodology one also requires the knowledge of the rotorcraft noise sources in different flight regimes. In the end one must also be able to validate the theoretical noise predictions with test data. The signal restoration methodology is a portion of this methodology development process because it provides a vital link between the test and predicted data. In addition to improving the present acoustic data base, the signal restoration simplifies the flight testing because it dispenses with a need to acquire flight data with another aircraft flying with the rotorcraft. The signal

restoration provides a method for restoring data acquired by the ground based microphones to free field data as if the microphone is moving with the rotorcraft.

In the measurement of the rotorcraft flyover noise, one encounters a situation where the receiver microphone is stationary and the source is in motion. The acoustic signal registered on the ground is highly transient in nature and because of this it falls into a class of signals which are called non-stationary signals. During the data reduction it is a common practice to process such a signal through a narrowband analyzer to estimate the spectral components of the signal. To analyze this signal certain assumptions about the temporal properties of the signal must be made because the analysis methods are those which are used for stationary signals. It is routinely assumed that the signal process is "locally" stationary (signal characteristics are constant during the sampling period). Then the statistical (ensemble) average is replaced by a "short" or "local" time average. However, in reality the signal is strongly distorted and modulated by the motion of the source relative to the receiver, so there is no logical justification for assuming that the signal is locally stationary. In the end we may ask: "What have we measured?"

The signal from the source, while propagating to the observer, undergoes several types of perturbations caused by the source motion, by the atmospheric effects, and by the interaction with the ground plane. Therefore in order to perform the total signal restoration all of the above mentioned phenomena should be addressed. In this report we shall limit our effort to removal of the kinematic effects, such as Doppler shift and the wave expansion effect, because these two mechanisms cause first order of magnitude perturbations.

For identification of the rotorcraft noise sources, typically a narrowband analysis of the flight data is performed. Because the rotorcraft noise spectrum is dominated by tones, de-Dopplerized narrowband spectra are essential for correctly identifying the tones and estimating their level at various observation angles.

The narrowband analysis, without de-Dopplerization, produces incorrect frequencies and tone levels in the spectrum due to the smearing effects of the Doppler shift. In some cases, investigators have resorted to a relatively wide analysis bandwidth, 25 or 50 Hz, to reduce the smearing effect; however this has lead to the conclusion that tones do not exist at frequencies above 200 Hz and that broadband noise is dominant. This conclusion has been shown to be incorrect for most performance conditions based on a narrowband analysis of the data from several wind tunnel experiments.

Complete signal restoration offers two distinct advantages: first, the tones become sharp, giving greater resolution; and secondly, their frequencies and levels are correct, allowing positive identification of the harmonics, side bands and interaction tones. It also becomes possible, for example, to plot a field shape, showing the level of a particular tone as a function of its emission angle.

The work in the area of signal restoration has been carried on by many researchers in various disciplines where non-stationary data is present. In this report we shall discuss work of several researchers which is applicable to the problem at hand. The discussion is intended as the background material for a better understanding of the processing of non-stationary signals.

The methods formulated to extract information from a non-stationary signal have been addressed by several researches. While there has been considerable activity in describing the non-stationary data in various ways, the problem of characterizing the "time variable spectrum" produced by the process has been rarely discussed. The approaches taken to restore the data, or to estimate the source parameters, can be broadly separated into the time and frequency domain methods. The two domains can be connected via Fourier transform so that if the data is in the frequency domain, an inverse transform will yield the time domain.

Previous work on de-Dopplerization has been limited by the hardware and not conceptual difficulties. The limitations of the data processing equipment (such as A/D converter speed and resolution, high speed data transfer from the computer memory to a disk or tape, and the requirement to handle vast amounts of data) have previously impeded the use of the computers for de-Dopplerization of non-stationary data.

Narrowband analysis and de-Dopplerization of flyover data have also been performed for both a simple acoustic source (Mueller and Preisser [1]) and a small turbofan engine (Preisser and Chestnutt [2]). In both of the cases the data is compared to narrowband static data. The software description is available in reference [3]. The de-Dopplerization was performed simply by adjusting the frequency scale in the spectral domain. This technique does not eliminate the smearing of tones, caused by the continuously changing Doppler shift, although the result is improved to some extent by taking short averages and employing an ensemble averaging technique with multiple microphones. A more sophisticated approach to the signal restoration is a so called "evolutionary spectral density" concept which was developed by Priestley [4]. This is spectral model that admits time and frequency, which results in a "moving" spectral density and which has properties that seem applicable to analysis of the non-stationary signal. The

spectral representation for a non-stationary stochastic signal can be represented by an integral over the frequency range of a product of modulation weighting functions with a sine and cosine series.

This concept has been further advanced by Tsao [5], [6] and Hammond [7]. The computational methodology depends on the synthesis of a time varying shaping filter which will restore the amplitude and frequency modulation effects. The method is theoretically sound but somewhat computationally difficult to implement. For each instance in time (each sample block) a set of filters must be synthesized for each type of modulation.

Another approach to the estimation of the source function from the field is by application of the Lorentz transformation for time scale transformation. This approach is described by Norum [8], Maestrello [9], and Chow [10].

More recently, Howell et. al. [11] have performed de-Dopplerization on the aircraft flyover noise in the time domain, while Verhas [12] has performed almost a complete signal restoration in the time domain. The two methods are conceptually the same in the de-Dopplerization part but differ only in implementation of the process. Howell et. al. [11] used an adaptive sampling technique to achieve de-Dopplerization. This method is similar to order tracking methods used in rotating machinery spectral analysis when the speed is changing. From the source position vs. time, the source velocity and the sound propagation vector direction are computed. The required sampling rate is a function of the source and propagation vector velocity.

On the other hand Verhas [12] has used a constant sampling rate to digitize the signal. He then time shifted the data in proportion to the amount of Doppler shift. This approach has been selected for implementation to perform the de-Dopplerization portion of the signal restoration. The reason for selecting this method is that digitizing equipment with a fixed sampling rate is generally available, while equipment with a variable sampling rate is not as commonly available. The variable sampling rate method has a draw back in that each data track must be processed separately, because each has its own particular sampling rate. On other hand, the fixed sampling can be performed in parallel on all the data tracks and adjusted for differences in the restoration process.

3.0 DEFINITION OF NON-STATIONARITY AND ITS EFFECTS ON SOUND PROPAGATION

The signal is defined to be non-stationary when its properties become a function of time or space and are not repetitive within the observation window. The non-stationarity may be modeled as a

modulation process of a stationary signal. The stationary signal may be modulated in frequency, amplitude or as a combination of frequency and amplitude at the same time. Rotorcraft noise, as observed by a fixed receiver, is a non-stationary process because the stationary source undergoes amplitude and frequency modulation.

As the sound propagates from a moving source it is affected by the environment through which it propagates or boundaries at which it makes contact. The source motion, the atmosphere, wind, turbulence, and ground are all dynamic phenomena, that is, they are all functions of time or space. Because they are a function of time or space, their interactions with the signal cause the signal to be non-stationary. The effects of this phenomena will be evaluated and considered for inclusion into the signal restoration process.

Since the above stated phenomena are dynamic in nature, the net effect on the data can be characterized by amplitude and frequency modulation due to the above mentioned mechanisms. The source motion causes the frequency modulation known as Doppler shift, while the change in distance between the receiver and the source (wave expansion) cause the amplitude modulation.

In this report the method of ray acoustics will be used to derive various relationships required for signal restoration. It is felt that the theoretically more correct methodology of wave acoustics is an unnecessary complication at this stage of restoration methodology development. It should be noted that, in the case of rotorcraft testing, the propagation distances are short (on the order of 1 km or less) and the environmental effects, wind and temperature, are restricted to a reasonable window.

3.1 Frequency Modulation

Frequency modulation is defined as a change in the frequency spectrum of a signal with time. The frequency modulation of the emitted signal from a moving source, as observed by a fixed receiver, is caused by the source motion and motion of the atmosphere through which the sound is propagating to the observer. It should be noted that the atmospheric motion will result in a Doppler shift only in the presence of wind or temperature gradients along the propagation path.

The frequency modulation mechanisms which have been identified are: Doppler shift due to the source and medium motion, 3.1.1.1, Rotor speed shift, 3.1.1.2, and Doppler shift temperature gradient, 3.1.1.3. Modulation due to source motion can be computed from the source position coordinates vs. time while medium motion is usually measured during the test with a meteorological instrumentation package attached to a tethered balloon.

3.1.1 Doppler shift due to source and medium motion

The relative motion between the source, medium and the receiver cause frequency modulation which is commonly called Doppler shift. The Doppler shift is shown schematically in Fig. 1. The relationship between the emitted source frequency and observed frequency is give by:

$$Fs = Fo\,[1 - \{Vsg \cos(T)/(C + Vwn)\}] \qquad [1]$$

where:
- Fs = frequency of the source
- Fo = frequency at the observer
- Vsg = source velocity relative to ground
- T = angle between the propagation ray and flight path
- C = speed of sound
- Vwn = component of wind speed along propagation ray

3.1.2 Frequency shift due to the change in rotor speed

A frequency modulation process which is not due to any propagation effects is modulation due to changes in the rotor speed. The changes can be significant since variations of +/- 2% have been observed during the flights. For a typical main rotor speed of 5 Hz for a 4 bladed rotor, a 2% change is equivalent to 0.4 Hz for the first harmonic, but 8 Hz for the 20th harmonic. If the analysis bandwidth is 1 Hz, then this will cause a significant broadening of the 20th harmonic peak.

The correction for the rotor speed change can be expressed as

$$Fs = Fr\,(RPMo/RPMr) \qquad [2]$$

where:
- Fs = Source frequency
- Fr = Reference frequency
- RPMo = Measured rotor speed
- RPMr = Reference rotor speed

3.1.1.3 Doppler shift due to temperature gradient

The effect of the temperature gradient along the propagation ray is to change the local speed of sound; hence, the effect can be accounted for by computing the local speed of sound and integrating the effect over the propagation path. The details of the correction can be found in reference [13].

3.2 Amplitude Modulation

The amplitude modulation is defined as a change in the amplitude of the signal with time. The amplitude modulation of the emitted

signal from a moving source, as observed by a fixed receiver, is caused by the source motion, absorption and scattering of the sound by the atmosphere, as it propagates to the observer.

The constantly changing distance between the source and the receiver causes the change in the propagation length and hence the change in the amplitude due to the wave expansion effect. The amplitude modulation effect is shown schematically in Fig. 2.

As the sound is propagated through the medium, the energy contained in the ray is being absorbed by the vibrational relaxations of the oxygen and nitrogen molecules in the air. The relaxation process is a result of the nontranslational mode of energy storage within the molecules that requires a finite time to adjust to changes in the translational energy, as expressed by the temperature. This adjustment time introduces a lag in the sound wave between changes in the pressure and density and results in attenuation of the amplitude of the acoustic wave.

3.2.1 Wave expansion effect

The most common wave expansion effect is the spherical spreading from a point source. This type of propagation mechanism will be used for signal restoration.

$$dL = 20\log(Ro/Rr) \qquad [3]$$

where: dL = change in the sound pressure level in dB
Ro = distance at the emission time to observer
Rr = reference distance.

3.2.2 Doppler Amplification Effect

The relative motion between the source, and the receiver causes amplitude modulation which is commonly called Doppler amplification. The Doppler amplification is given by:

$$dL = 20\log([1- M\cos(T)]^{-1}) \qquad [4]$$

where: dL = Change in sound pressure level in dB
M = Source Mach number
T = Angle between the propagation ray and flight path

3.2.3 Atmospheric Absorption

The sound pressure decreases exponentially during propagation through the atmosphere due to the atmospheric absorption effects. The absorption coefficients are computed in accordance with ANSI S1.26-1978 standard [14]. The relationship can be expressed as :

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$$dL = 8.69\ A\ R \qquad [5]$$

where:
- dL = Change in pressure level in dB
- A = Absorption coefficient nepers/m
- R = Propagation distance in m.

This relationship is valid for a homogeneous, still atmosphere of normal composition. That is, pressure, temperature and molar concentration of water vapor can each be specified by a single fixed number. This condition is very rare in the real world because the above mentioned parameters are a function of propagation distance.

3.3 Assessment of Non-Stationarity Effects on Sound Propagation

In this section we shall briefly discuss the effects which will be included in the restoration as well as those which are omitted. The mechanisms, which are not a part of restoration process at this stage of the development, will be evaluated as to their effect on the final results.

3.3.1 Frequency Modulation Effects

3.3.1.1 Doppler shift due to source motion

The Doppler shift is a result of the source motion relative to the observer. This effect has a major effect on distorting the wave front as it propagates from the source to the observer. This effect will be fully accounted for in the signal restoration process.

3.3.1.2 Doppler shift due to medium motion

The motion of the medium relative to the observer causes the Doppler shift as if there is a velocity gradient between the source and the observer. If on the other hand the medium is moving at a constant speed but the observer and the source are at rest, as in the case of wind tunnel testing, the Doppler shift does not occur. The Doppler shift, due to medium motion, has a smaller effect on the wave front distortion as compared to the source motion. The effect is proporational to the wind velocity vector along the propagation ray. Acoustic testing of rotorcraft is restricted to days when wind with horizontal velocity is less than 5 m/sec. Assuming a linear wind gradient, it is estimated that a 5 m/s wind at the ground level will be 15 to 20 m/s at 150 m flyover altitude, the Doppler shift is estimated to be about 1.5% at small elevation angles and decreases with increases in the elevation angle. The estimate is based on Roy [13].

Even though the effect is small in terms of the overall Doppler shift this effect should be incorporated into the restoration procedure in the second stage. The correction should be based on the measured wind velocity profile at the time of the test.

3.3.1.3 Doppler shift due to temperature gradient

The temperature gradient between the source and the observer also causes Doppler shift. The shift has a smaller effect on the wave front distortion as compared to the source motion. Basically the effect is proportional to the change in the local speed of sound along the propagation path.

Acoustic testing of rotorcraft is usually carried out in a temperature range of 2 to 35 degree C. Using a typical temperature lapse rate of 0.03C/m, the temperature is estimated to be 4.3 deg less at 150m, than at ground level. The estimate of the temperature near the ground is also important because of the steep gradient, usually present near the ground. For estimating the Doppler shift an increase of 2 degrees of the ground temperature above the air has been assumed. Using a 6.3 degree differential in temperature and using the method of [13], it is estimated that the Doppler shift will be about 0.3% at a 20 degree elevation angle and will decrease with an increase of the elevation angle.

The effect of temperature gradient, while relatively small, should be included in the future extension of the restoration methodology. These corrections should be based on the measurement of the temperature profile.

3.3.1.4 Frequency shift due to rotor speed shift

The changes in the rotor speed during the flight due to the action of the engine controls results in a frequency modulation. This shift will be fully compensated for in the restoration procedure if the time history of rotor speed is available.

3.3.2 Amplitude Modulation Effects

A second series of modulation effects arise due to the source motion and are grouped here as amplitude modulation effects. The amplitude modulation effects are due to many mechanisms, but only the wave expansion and Doppler amplification effect will be included in the restoration procedure. The other effects such as ground effect, refraction, atmospheric absorption, and turbulence will be discussed as to their effects on the amplitude modulation.

3.3.2.1 Wave expansion effect

The motion of the source relative to the observer causes a continuous change in the propagation distance and, as the result, causes a constant change of the received signal amplitude. In the far field, signals are considered to be spherically spreading wave fronts from a point source, giving an attenuation rate of 6 dB per doubling of distance. As we move closer to the source the point source approximation becomes less valid and for the helicopter the main noise source becomes a rotating, distributed source with finite dimensions. For the purpose of the signal restoration a simple monopole source and attenuation rate of 6 dB per doubling will be assumed. The effect will be fully compensated for in the restoration process.

3.3.2.2 Doppler amplification effect

The relative motion between the source and the receiver causes amplitude modulation, which is commonly called Doppler amplification. The Doppler amplification will be fully accounted for in the restoration procedure.

3.3.2.3 Ground effect

The presence of a ground surface boundary adds a reflected signal to the direct signal, and the resultant phase shift between the two signals creates an interference pattern at the receiver. The interference pattern introduces an amplitude modulation effect. This modulation is a function of frequency for microphones mounted above the ground surface, but is a constant (6dB) for ground plane microphones over a hard surface. The signal restoration, as presently formulated, assumes that the data has been acquired with a ground plane microphone.

The interference of direct and reflected waves depends on the frequency, the geometry of the source and receiver positions (i.e. the height above the ground and the separation distance) and the ground impedance. The interference pattern will be varied by topographical changes, by refraction due to heterogeneousness and by turbulent scattering.

The frequency dependent constructive addition of the signals, in the presence of a hard reflective plane, can add as much as 6dB for coherent signals and 3 dB for incoherent signals. Destructive addition can, theoretically cause the complete destruction of the signal, although this is not practically possible. By placing the receiver in plane with the ground surface only constructive addition occurs at most frequencies of interest.

A finite ground impedance alters the phase and amplitude of signals, thereby altering the interference pattern. The result is that the maximum reinforcement is less than the 6dB (typically 2 to 4 dB) due to ground absorption and the cancellations are not as intense and sharp (Yoerkie [21]) as for the hard surface case.

The interaction of a non-planar wave with the ground leads to a more complex mathematical model which requires the creation of a ground and a surface wave. The model for these waves was developed as a direct analogy to electromagnetic wave propagation (the Weyl-Van der Pol solution of the wave equation). The reason for this more complicated model is the need to match the variation in curvature of the wave fronts with the distance along the boundary. Physically, the surface and ground waves are not well understood. It is believed, however that for the source and the receiver on the ground, (when the direct and reflected waves should cancel completely), the noise propagation is primarily due to the ground and surface waves.

In general, the ground and surface waves will significantly affect low frequency signals when the source and the receiver are near the ground, and at long separation distances. The surface wave, at relatively short distances will cause a 3 dB per doubling of distance spreading. At further distances the ground wave dominates and gives 6 dB more per doubling of distance than spherical spreading. The result is a 12 dB reduction per doubling of distance at some distance from the source (dependent on geometry, frequency and ground impedance).

An idealized representation of losses due to spherical spreading and ground effects (without the interference pattern) is shown in Fig.3.

3.3.2.4 Refraction effects

The atmosphere is not isothermal under normal conditions, yet most acoustic propagation models make this assumption. The temperature and wind heterogeneity can occur as random fluctuations from the convection cells and turbulence, or as slow diurnal variations in the atmosphere immediately adjacent to the ground surface. During the daylight hours the atmosphere is typically warmer near the ground than above. The strong temperature gradients tend to extend for several meters above the ground where most microphones are located. In particular, the ground plane microphone, which is used to avoid the ground plane effect, is unfortunately located so as to receive the maximum effect of the refraction due to the temperature and wind gradients.

The result of the temperature and wind velocity gradients is a sound velocity gradient. Increasing sound velocity with height gives refraction (or bending) of the sound rays downward, making possible multiple ground reflections. Decreasing sound velocity with height causes refraction of the sound rays up into the atmosphere. More complex sound velocity profiles will cause wave guides and acoustic mirage types of phenomena.

Refraction can cause focusing of sound, forming caustics or defocusing of sound resulting in acoustic shadow zones. The theoretical limits are complete cancellation or infinite noise levels. These extremes are both physically impossible, however, 30 dB fading is not unusual. The effects of the refraction are most pronounced at the low elevation angles. It has been reported in the literature that a variation of +/- 8 dB can be expected at the ground plane microphone. Under some unusual conditions near total attenuation can occur (Van Moorhem [16]).

3.3.2.5 Atmospheric absorption

Atmospheric absorption is an amplitude modulation effect because it is a function of distance. The atmosphere can be modeled as a homogeneous or stratified (the later is closer to the reality and should be used whenever the data is available).

Classical atmospheric absorption is the result of shear viscosity, thermal conductivity, mass diffusion and thermal diffusion in the atmosphere. The current ANSI standard (ANSI S1.26-1978), "Method for the Calculation of the Atmospheric Absorption of Sound by Atmosphere [14], includes the effects of rotational relaxation of air and vibrational relaxations of nitrogen and oxygen. Recently, Zuckerwar [15] has proposed changes to the standard based on more accurate measurements of the relaxation phenomena giving improved low frequency results.

Although the current ANSI version is the best available standard for estimation of the atmospheric absorption it tends to under estimate the absorption coefficients at low frequencies. Therefore Zuckerwar's correction should improve rotorcraft low frequency source predictions.

3.3.2.6 Turbulence effects

Turbulence has three major effects on propagating acoustic signals: fluctuations of amplitude and phase, attenuation due to beam broadening (or scattering), and refraction due to large scale turbulent eddies.

Fluctuations of phase and amplitude (small scale turbulence) have been shown by Embleton et. al. [17] to cause very small differences in the signals except in the region of interference minima (where slight changes in phase can cause the cancellation of sound to be less efficient).

Attenuation of the sound due to beam broadening has shown small variations due to scatter, since the scatter is mostly in the forward direction. Clifford and Brown's model predicts this attenuation well. However, forward scatter is an issue when refraction causes a shadow region. Under these conditions, turbulent scatter of sound will penetrate into the shadow zone.

Refraction by large scale eddies causes significant variation in acoustic signals. When the path length is less than about 200 meters these large scale eddies can surround the entire propagation path and cause an effective temporary change in the microclimate.

3.4 Proposals for Further Extension of Signal Restoration

Modeling all of the atmospheric and ground interaction phenomena is a monumental task. Therefore, evaluation of the feasibility and incremental effect on accuracy needs to be addressed.

Atmospheric absorption is well defined and should, therefore, be included in the next version of the signal restoration. Two forms, ANSI and Zuckerwar's extension of the ANSI standard should both be included. This will provide only a small improvement over short distances, but is more significant over longer ranges. The implementation of the atmospheric absorption could be in a form of inverse filtering applied to the signal. A series of filters will be synthesized, which in the frequency domain, correspond to the atmospheric absorption spectrum. The filters are unique for a given atmospheric condition and are computed only once. This seems to be an efficient method to remove the atmospheric effects from the time domain data.

The restoration in the frequency domain, as is now frequently done, requires computation of the FFT for each instant in time, compensating for the absorption and then performing the inverse FFT to return to the time domain. The absorption transfer function is unique for a given atmospheric condition and is computed only once. The FFT operations is carried out on blocks of data 2048 points long. Thus, a typical data set of about 500,000 points would require 250 FFT and inverse FFT operations. Although, both methods are computationally feasible, the digital filter approach is more direct and computationally more efficient.

Small scale turbulence does not cause large variations in signal (maximum of 2 to 4 dB variation). The losses associated with this phenomenon are well defined by the Clifford and Brown model, which can be added following the atmospheric absorption.

At this stage, in order to avoid the problem of the interference pattern associated with the coalescence of direct and ground reflected waves, the receiver is placed in-plane with the ground surface. This causes only constructive interference (sum of the signals) for the frequency range of interest. However, the ground impedance as a function of frequency would need to be known to accurately assess the amount of signal increase, (the limit is +6dB for a perfectly reflecting surface). Since most surfaces are "hard" at low frequencies, where our interest lies, the +6 dB approximation will be assumed. The inclusion of the ground reflection effects for microphones above the reflecting plane is not proposed at this time.

It should be noted, for ground plane microphones, that one pays a penalty in terms of the refraction effects. An uncertainty of +/- 8 dB is possible (but has a low probability) due to the refraction effect near the ground. Since the refraction effects are most significant near the ground surface, it is possible to model the effect provided the temperature and wind profiles are known during the data acquisition. The refraction should also be considered for future inclusion into the restoration procedure.

In order to further validate the restoration procedure a test should be conducted using other techniques of flying over a microphone array and then de-Dopplerize this data in the frequency domain. Then comparison between the two methods could be made.

4.0 RESTORATION METHODOLOGY

The signal restoration is a technique used to transform a non-stationary signal to a stationary one. Methods of the stationary signal analysis can than be used to extract desired information from the signal. The methods used to deal with various modulation mechanisms will be described in detail. The algorithms used for signal restoration will be described under the heading of the Restoration Software 5.0.

The restoration methodology can be divided into two parts, restoration of the frequency modulation and restoration of the amplitude modulation effects. The restoration process is depicted schematically in Fig. 4. The restoration methodology is based on the separation of the modulation effects so that the principal of superposition can be applied. Therefore each modulation effect can

be compensated for independently of the others. For purely computational purposes we have chosen to perform the amplitude restoration first, followed by the frequency restoration.

4.1 Amplitude Modulation Restoration

The amplitude modulation is caused by a constant change of the distance between the source and the receiver, Doppler amplification, atmospheric effects, source directivity, and ground effects. The nature of the modulation process is shown graphically in Fig 2. The source, S, is moving along a path at a constant speed, v. The receiver at a fixed position, R, sees a moving source along line, SR, which is characterized by a continuously varying angle, T. The signal at the source, S, is As(t) and the signal at the receiver, R, is Ar'(t'). Note that the primes are used to denote the non-stationary variables. For a monopole source when T is 90 degrees, Ar'(t') will be at the maximum.

If the receiver, R, were moving at a constant speed, v, on a path parallel to the source path, (so that angle, T, is held constant), then the signal at R will be stationary. The sound pressure level at R will then be a function of separation distance, SR, and emission angle, T.

If we fix the relative positions of the source, S, and receiver, R, and if the source, S, generates a stationary signal, As(t), then a stationary signal, Ar(t), will be received at R. In order to obtain a non-stationary signal, Ar'(t), as an output at R, signal, As(t), must be transformed in the way a moving source would do it. However, the direction of the transformation in the restoration process is reversed, we take a non-stationary signal and compute a stationary signal. This transformation implies compensation for: (i) the distance variation between the source and the receiver during the flyover. (ii) Doppler amplification, (iii) the atmospheric effects, and (iv) ground effects. The restoration for the first two effects will be implemented in this version of the restoration process while the next two effects are left to be included in the future, as is discussed elsewhere in this report.

4.1.1 Wave Expansion Effect

The first transformation is compensation for the wave expansion effect. The restoration is based on the ray acoustics formulation. It is assumed that the source is in the far field and the inverse square law is applicable. If the experience shows that the ray acoustics model is inadequate, then a more complicated wave expansion formulation can be implemented.

Compensation for the wave attenuation due to the wave front expansion is:

$$As(t) = Ar(t)\ [l(t)^2/lr^2] \qquad [6]$$

where: $Ar(t)$ = Amplitude of restored signal
$As(t)$ = Amplitude at the source
$l(t)$ = Propagation path length
lr = Reference distance

The procedure depends on the knowledge of the propagation path length for each acoustic data point. For example if the acoustic data were sampled at a rate of 10 kHz then the position data must be also measured at the same rate. In general, the position tracking instrumentation is not capable of directly providing position data at such a high rate. Therefore, it is necessary to compute the propagation path by interpolation of the position data.

During the rotorcraft testing, the position data is usually available at a rate of 2Hz. The interpolation of the data does not pose a serious problem because the rotorcraft flight dynamics preclude abrupt changes in the flight path.

At this point, it is worth while to point out that the propagation distance is not generally measured directly during the flight test, but it is computed after the test. Under usual test circumstances the acoustic and the position data are acquired by separate data acquisition systems. After the test, data is synchronized via the time code, which has been recorded during the test by both systems. What is known is the rotorcraft position verses time and the time of signal reception, rather than the rotorcraft position at the signal emission time. The rotorcraft position at the signal emission time is then computed from the position data. In order to simplify the computation of the rotorcraft position at the emission time, the assumption is made that the rotorcraft velocity was constant during the signal propagation time. The assumption of the constant velocity is usually sustained in the flight test, since the pilot is controlling the speed. During flight testing, typical speed changes on the order of +/-.0.05 m/s have been observed.

4.1.2 Doppler Amplification Effect

The sound radiation from a moving source is affected by the source motion itself and results in the amplitude modulation of the signal at the receiver. This amplitude modulation process is known as Doppler amplification, because it is proportional to the Doppler shift. The Doppler amplification correction, based on the work of Morse and Ingard [18], is:

$$As(t) = Ar(t)/(1- M \cos T)^2 \quad [7]$$

where: $Ar(t)$ = Amplitude of restored signal
$As(t)$ = Amplitude at the source
M = Mach number
T = Angle between the propagation path and flight path

4.2 Frequency Modulation Restoration

The frequency modulation is caused by the source motion, changes of the main rotor speed during the flight and presence of temperature and wind velocity gradients. The current restoration procedure will address only the Doppler shift and the rotor speed changes. Other frequency modulation mechanisms will not be included in the restoration procedure at this time. The frequency restoration procedure is shown schematically in Fig 5.

4.2.1 Doppler Shift Restoration

The restoration of frequency modulation caused by Doppler shift is based on computation of the Doppler shift. The restoration methodology is base on sampling the acoustic signal at a constant sampling rate and then time shifting the signal in proportion to the Doppler shift. When the data is interpolated back to the original sampling rate the resultant signal is a de-Dopplerized one. The time shifted period is given by:

$$Pr(t') = Po(t) \ / \ (1- M \cos T) \quad [8]$$

where: $Pr(t')$ = Restored data sampling period
$Po(t)$ = Nonrestored data sampling period
M = Mach number
T = Angle between the propagation path and flight path

At this point one may select a new sampling period for the de-Dopplerized acoustic data. The new period can be equal to the original one or longer than the original sampling period. If the data is resampled at a shorter period than the original, as may be in the case where one desires to increase the frequency resolution, wrong results may occur because of the aliasing phenomenon.

4.2.2 Rotor Speed Shift Restoration

The main rotor speed changes cause frequency modulation. During flight tests, changes in rotor speed of +/- 2% have been observed. This amount of the rotor speed shift can result in a significant amount of smear in the frequency domain. Therefore, compensation for the main rotor speed variation, is based on normalization of

the rotor speed. The restoration, as in the case of the Doppler shift is based on changing the sampling period of the data. In the restoration procedure the period of the time shifted data is the sum of the Doppler and rotor speed shifts.

$$Pr(t') = Po(t)(Rn/Rm(t)) \quad [9]$$

where: $Pr(t')$ = Restored data sampling period
$Po(t)$ = Nonrestored data sampling period
Rn = Nominal rotor speed
$Rm(t)$ = Measured rotor speed

5.0 SIGNAL RESTORATION SOFTWARE

The signal restoration is implemented on a digital computer because of the ease with which data can be manipulated. To perform any type of computation on a computer, programs must be written to operate the computer. In this section we shall discuss the software design philosophy, computer, operating system attributes, software portability issues and software documentation.

5.1 Software Design

A structured programing philosophy was used to design the signal restoration software. This software design approach was followed as far as it is practical within the confines of FORTRAN 77. FORTRAN 77 does not fully lend itself to structured programing, as does C or Pascal. The structured programing uses a "top down" design methodology for software. The top down design progressively breaks down the functions until each independent function has been identified. Thus an analyst can identify and optimize any inter-functional relationships that are discovered in the breakdown process. Most of the design and logic problems can be discovered as the design is progressing, although some will remain undiscovered until the unit debugging and testing.

It is of paramount importance, in structured programing, that the design always proceed from top down. This progression begins with an overview of the major components of the system without entering into the details. Each major component is then broken down into subcomponents, with an emphasis on what is done, and not particularly when.

At this point it is usually appropriate to present the information in a form of a chart showing the flow in the logic process (Fig. 6). This process of going from function to subfunction continues until the programing language level is reached. Identical, or somewhat related, functions are identified so that the code can be

reused. The program structure must be functionally organized so that any design changes to one portion of the program affect only the revised area.

The signal restoration program is divided into a set of functional modules, so that each module performs a particular restoration process. This approach permits simultaneous coding of the program, and testing. Since each module has its own I/O files, it can be tested independently of other modules. As always, there is a drawback, that one may have to create a test data set prior to one being available. The structural programing approach will facilitate the installation of the signal restoration software on different computers and simplify the software maintenance.

The program is designed to be operated in an interactive mode during the input and postprocessing operations. The input portion of the program is set up as a dialog, where the user is prompted to enter the appropriate data or answer simple questions. If a batch operation is required, then certain modifications to the program would be required. The postprocessing operations, such as time or frequency domain functions and plotting, are interactive.

5.2 Computer, Operating System and Programing Language

The software was developed on the Hewlett Packard HP 9000/550 minicomputer. This computer provides full 32 bit internal and external data paths, 32 and 64 bit mathematics (IEEE floating point format), 32 bit memory addressing and virtual address space of 500 Megabytes. The computer is operated under HP-UX operating system which is the Hewlett Packard implementation of the A.T.&T. UNIX System V, release 2.3, with some UCB UNIX 4.2 BDS and Hewlett Packard enhancements. The compiled software will run under most other UNIX operating systems without any difficulties.

The language used for coding the software is FORTRAN 77 (ANSI X3.9-1978) with MIL-SPEC 1753 Military Standard FORTRAN extensions. Hewlett Packard extensions to FORTRAN 77 were not used because they are not a part of any recognized standard. The FORTRAN 77 used on this particular computer has a working space of 512 kilobytes. When the program exceeds this limit, the program must include virtualizing compiler directives.

5.3 Software Portability Issues

The program is coded using standard FORTRAN 77 (ANSI X3.9-1978) computer language with MIL-SPECI 1753 Military Standard FORTRAN extensions. However, as with any system, some local dialect variations are inevitable. The Hewlett Packard version of FORTRAN 77 does not have dynamic memory allocation without resorting to the C language routines. Therefore, the dimensioned variables must

have full dimensions in programs and subroutines. Because the program handles large arrays, program requirements exceed the limits of the allocated user memory block, 512 kilobytes. Hence, virtualizing instructions are necessary for the arrays in common statements in order that the program may run.

Aside from the above, programs will need very little recoding for installation on other computers. The graphics output routines are coded in C, using the Starbase Graphics Library which is a part of HP-UX system. The majority of graphics commands are based in GKS kernel, and follow the current ANSI Computer Graphics Interface standard. The graphics routines are provided for reference purposes.

The file system, structure in HP-UX, is hierarchical, which is a "treelike" structure allowing users to organize files in a convenient and logical fashion. The file naming conventions are based on existence of the directories under which groups of files have been stored. FORTRAN 77 language does not recognize the difference between the directory files and data files, therefore the programs which are coded in FORTRAN do not use the directory feature. However, this results in unnecessarily long file names on the system which do not support hierarchical file structure.

5.4 Program Documentation

The software documentation, as well as the software development, adheres to the structured approach. The program modules and subroutines are self documented within the code. Each module has a header, which contains the file information, subroutines called and a listing of all the variables used in a particular section of the program. The software documentation is divided into two parts: 5.4.1, Description of the module's purpose, algorithm, and computational methods used for implementation, and 5.4.2, Description of the data and file structures, used for input data and output data.

5.4.1 Source Code Documentation

The source code documentation describes various algorithms used for computation and explanation of the input and output variables.

5.4.1.1 Module 'INPUT'

The function of the 'INPUT' module is to prepare the "input" file which contains the global type of data, and which is used by many of the computational modules.

5.4.1.1.1 Data input

The user is prompted to input data and answer questions as they appear on the screen. The program attempts to screen as many input errors as possible, at the source line. However, it is not always possible to screen out all of the errors, thus, at the end of the input and prior to writing to the file, the user is requested to confirm the validity of the input.

5.4.1.2 Module 'RETARDTIME'

The function of the 'RETARDTIME' module is to compute the rotorcraft position and velocity at the emission time, so that the sound propagation path length and the emission angle can be computed. The propagation path length, Mach number and emission angle are used in frequency and amplitude restoration procedures.

5.4.1.2.1 Computation of the angle between the flight path and a line between microphone and rotorcraft at received time

The first process is computation of the angle between the flight path and the propagation ray at the reception time. The flight path geometry is shown in Fig. 7. The equations used to compute the direction cosines of the flight path vector and the line connecting the microphone position and rotorcraft position are shown below.

$$(A1) = Xrr - Xr \qquad [10]$$

$$(B1) = Yrr - Yr$$

$$(C1) = Zrr - Zr$$

$$(A2) = Xmr - Xr$$

$$(B2) = Ymr - Yr$$

$$(C2) = Zmr - Zr$$

$$R1 = [(A1)^2 + (B1)^2 + (C1)^2]^{\frac{1}{2}} \qquad [11]$$

$$R2 = [(A1)^2 + (B2)^2 + (C2)^2]^{\frac{1}{2}}$$

$$\cos(T) = \frac{(A1)(A2) + (B1)(B2) + (C1)(C2)}{R1 \quad R2} \qquad [12]$$

where:	A1, B1, C1	= direction cosines of flight path
	A2, B2, C2	= direction cosines of line from rotorcraft to microphone
	Xr, Yr, Zr	= Coordinates of rotorcraft at receive time
	Xrr, Yrr, Zrr	= Coordinates of rotorcraft at receive time plus offset
	Xmr, Ymr, Zmr	= Coordinates of microphone position
	R1	= Distance between receive position and offset position
	R2	= Distance between receive position and microphone position
	T	= Angle between the R1 and R2

In order to compute the direction cosines of the flight path the usual procedure would be to take the next rotorcraft position coordinate. However, this presents a computational problem. The rotorcraft moves about 5 cm between the successive increments in time while the distance between rotorcraft position and microphone is several orders of magnitude larger, typically 100 to 1000 m. The disparity between the two values of R1 and R2, cause a problem in the computation of the angle T. To overcome this computational problem, an offset is introduced in order to increase the value of R1. The amount of offset selected is 217 data points, or 8 to 10 m. The data offset may be changed if the position sampling rate is reduced or the rotorcraft moves at a lower speed. For the results of this computation to be valid, it is assumed that the rotorcraft is not accelerating during the period of time equal to the offset. The assumption should generally be valid in view of the rotorcraft flight dynamics and the steady state conditions.

5.4.1.2.2 Smoothing of angle, T, data

The second operation is to smooth the angle, T, computed in the previous module. The smoothing of the data is necessary because the position coordinate data contain certain residual effects of the interpolation process, such as oscillation and possible discontinuities at the boundaries of interpolation segments. The smoothing method selected is based on passing a polynomial through a number of points to be smoothed and then computing the difference between the input value and the interpolated value. This difference is called "energy". After the energies of all the points have been computed, the points with the largest amount of energy is selected for smoothing. The amount by which the point is smoothed depends on its energy and the average energy in the region about the point. If the energy ratio of the point energy and the average energy of the points around the point is high, then the correction is equal to the full amount of energy. The

correction is added or subtracted as the case may be to the coordinate of the point. On the other hand if the energy ratio is low, then one half of the energy is used for correction. This process is repeated until the preselected level of smoothness is reached. During the software testing a smoothness of 0.2E-04, was chosen as a reasonable compromise between the smoothness and running time. For the test data set of 275,000 points 11 iterations are required to reach this degree of smoothness, and 10 seconds of CPU time. The user may select his own degree of smoothness depending on the preference or the residual roughness that can be tolerated in the end result.

5.4.1.2.3 Computation of velocity at receive time

The third process is the computation of the rotorcraft velocity at the reception time. The algorithm selected is based on the numerical differentiation using the central difference method. The algorithm is known as "averaged central difference", and has an error proportional to H^4 where H is the interval between the data points. In order to obtain a certain amount of averaging in time the interval, H, was doubled, so the velocity is computed over a slightly longer period in time.

$$V(I) = [X(I-2) - 8X(I-1) + 8X(I+1) - X(I+2)]/12H \qquad [13]$$

where: V(I) = Velocity at point I
X(I) = Coordinate of position in X direction
H = Interval between the coordinate points

The same equation is used to compute the velocity for Y and Z coordinates. The magnitude of the velocity vector is computed by the vectorial sum of the components.

5.4.1.2.4 Smoothing of velocity data

Following the computation of the velocity, the smoothing routine is applied to the velocity data for the same reasons as were stated previously for the angle, T. The degree of smoothness selected is 0.32 which is equivalent to variation of +/- 0.28 m/sec. in velocity. Again the choice of the smoothness value is a trade-off between the results and the running time of the routine. The smoother data results in higher degree of de-Dopplerization.

5.4.1.2.5 Computation of propagation path length and emission angle

The final computation is calculation of the propagation path length and the angle between the flight path and propagation ray. The propagation ray path length is given by:

$$RE = \frac{RS(M\cos(T')+[1-(M\sin(T')]^2}{[1-M^2]} \qquad [14]$$

$$T = T' - \sin^{-1}[M\sin(T')] \qquad [15]$$

where:
RE = Propagation ray path length
RS = Distance from microphone to rotorcraft at emission time
M = Mach number
T' = Angle between flight path and RS
T = Angle between flight path and RE

The values of RE, the propagation path length, is used for amplitude modulation restoration and the emission angle, T, is used in the frequency modulation restoration.

5.4.1.3 Module 'WAVEXPAND'

The function of the 'WAVEXPAND' module is to perform amplitude restoration caused by the wave expansion effect. The computation is based on ray acoustics and a simple point source. The assumptions are reasonable if the measurements of the sound pressure level are performed in the far filed. The atmospheric effects are not included in the amplitude restoration.

5.4.1.3.1 Amplitude restoration

The amplitude restoration is given by:

$$SPL' = SPL(RE/Ro)^2 \qquad [16]$$

where:
SPL' = Restored sound pressure level
SPL = Measured sound pressure level
RE = Propagation path length
Ro = Reference distance

The default reference distance for the rotorcraft is chosen to be 150m, which corresponds to the FAA/ICAO flyover altitude. For other flight profiles, such as approach, it should be changed to 120m to correspond to the standard measuring distance. In the takeoff case, the reference distance is rotorcraft dependent and normally varies between 100 to 180m. Other normalization distances can be substituted in the program as required by the user.

5.4.1.4 Module 'DOPPLER'

The function of the 'DOPPLER' module is to perform computation of the Doppler shift and Doppler amplification. Doppler shift is used in frequency modulation restoration while Doppler amplification is used in amplitude modulation restoration. These two functions are

combined into one module to save I/O operations because the variables used to perform the calculations are the same for both mechanisms.

5.4.1.4.1 Doppler shift

The computation is performed as a change in sampling period rather than as a calculation of the frequency shift. The usual formulation of the Doppler shift is used as it applies to the period. Since the frequency is inversely proportional to the period, the increase in frequency corresponds to the decrease in the period and vice versa. The computation performed uses this information to compute a new period which is proportional to the Doppler shift.

$$Dp = Do / [\ 1 - M \cos(T)] \qquad [17]$$

where:
- Dp = Period corrected for Doppler shift
- Do = Original period
- M = Mach number
- T = Angle between flight path and propagation ray

5.4.1.4.2 Doppler amplification

The amplitude of the sound emitted by the source is effected by the source motion. The amplification or attenuation of the sound is proportional to the Doppler shift and the emission angle.

$$SPL' = SPL/[1 - M \cos(T)]^2 \qquad [18]$$

where:
- SPL' = Restored sound pressure
- SPL = Sound pressure restored for wave expansion
- M = Mach number
- T = Angle between flight path and propagation ray

5.4.1.5 Module 'RPMSHIFT'

The function of the 'RPMSHIFT' module is to perform computation of the rotor speed shift. Rotor speed shift is used in frequency modulation restoration.

5.4.1.5.1 Rotor speed shift

The changes in the main rotor speed result in the frequency modulation. The purpose of this operation is to compute an increase or a decrease of the main rotor period relative to the reference rotor period.

$$Dr = DT(1 - RPMo/RPM) \quad [19]$$

where:
Dr = Time shift due to rotor speed shift
DT = Original data sampling period
RPMo = Reference rotor speed
RPM = Measured rotor speed

The computed rotor speed change correction is added to the Doppler shift.

5.4.1.6 Module 'TOTALSHIFT'

The function of 'TOTALSHIFT' module is to perform summation of Doppler and rotor speed shift. The total shift is used in frequency modulation restoration.

5.4.1.6.1 Total shift

The total shift is the sum of the Doppler and the rotor speed shifts.

$$Ts = Dp + Dr \quad [20]$$

where:
Ts = Total shift
Dp = Doppler shift
Dr = Rotor speed shift

5.4.1.7 Module 'RESTORE'

The function of the 'RESTORE' module is to perform frequency modulation restoration. The restoration process is started by time shifting the data and then performing interpolation with the original sampling rate or some other sampling rate which is lower than the original rate. The operation is performed on the sound pressure and on the emission angle.

5.4.1.7.1 Time scale

The first operation is to compute time shifted and restored time scales. The time shifted scale is computed with following algorithm:

$$T(I+1) = T(I) + TD(I) \quad [21]$$

where:
T = Absolute time of measured data in seconds
TD = Total time shift in seconds
I = Index 1,2,.... N

The restored time scale is computed in a similar manner.

$$Tr(I+1) = Tr(I) + DT(I) \qquad [22]$$

where: Tr = Absolute time of restored data in seconds
DT = Time increment of restored data in seconds
I = Index 1,2,.... M

Note that N and M are the number of points in the original array and the restored array respectively. The constraint is that M must be less than or equal to N.

5.4.1.7.2 Interpolation of sound pressure data

The second operation is interpolation of the sound pressure data to the new constant time interval. The Lagrange interpolation method was chosen because it is the most suitable interpolation routine for unequally spaced data points. The method is based on passing a nth-degree polynomial through n+1 pivotal points. In this particular case we have selected the 4th order interpolation polynomial, thus 5 points participate in the computation of the new data point. The details of the routine are given in Salvadori and Baron [18].

5.4.1.8 Module 'PSD'

The function of the 'PSD' module is to perform computation of Power Spectral Density (PSD) via Fast Fourier Transform (FFT), and to perform ensemble averaging of the PSD spectra. The program stores data in a form suitable for plotting.

5.4.1.8.1 Data conditioning

The first operation is computation of the number of ensembles which can be drawn from the time domain data set. Since the FFT routine uses 4096, the data must be divided into blocks of 4096 points.

5.4.1.8.2 Conversion from real to complex

The time domain data are real and must be converted to complex prior to entering an FFT computation.

5.4.1.8.3 FFT computation

The FFT routine computes the Fourier Transform of the equally spaced complex data using the Cooley-Tukey method. The details of the routine are described in Otens and Enochson [20].

5.4.1.8.4 Ensemble Sum

Following the FFT computation the magnitude of the spectral components is computed, and ensembles are summed.

5.4.1.8.5 Windowing of Data

Due to the nature of FFT computation, a "leakage" phenomenon is present in the spectra. In order to minimize leakage, spectral windows are used. However one pays a corresponding penalty, such as a reduced resolution and loss of orthogonality, which in turn results in a loss of degrees of freedom.

The program has three window options: Rectangular, Hanning, and GEO. The rectangular window is effectively no window, and offers the highest spectral resolution but also has the greatest amount of leakage. The Hanning window is a cosine function that is offset so that at the beginning and the end of data sample its value is zero. This satisfies the requirement that the date be periodic within the sample data block. The Goodman-Enochson-Otens, GEO, window was designed for application as a smoothing function in the frequency domain rather than as a tapering function in the time domain. The result is that the spectral window is 10% wider than rectangular window as opposed to 20% for the Hanning window. The details of the windows are given is reference [20].

5.4.1.8.6 PSD calculation

Following the windowing of the PSD data, the ensemble average of the spectra is computed.

$$PSD = 20\log(2ENS) + BW - CW \qquad [23]$$

where:
- PSD = Power spectral density in dB
- ENS = Ensemble sum
- BW = Band width correction factor
- CW = Window correction factor

5.4.1.8.7 Storage

Following the PSD calculation the array of spectral points is output to a file for subsequent plotting. The first 1600 data points are output, as it is a customary practice to cut off the upper 20% of data points because they may contain aliasing effects.

5.4.1.9 Module 'TIMEDOMMAIN'

The function of the 'TIMEDOMMAIN' module is to perform so called "time domain post operations" such as averaging, autocorrelation, and filtering routines. The program will store the output data in a form suitable for plotting.

5.4.1.9.1 Menu

This submodule is entered by selecting time domain post operations in the main menu. The program will then list a directory of all the runs under that job name and prompt the user for the run name it is to be operated on. After this, the program will read in the necessary files, show the start time, stop time, overhead time, and number of points in the data file. The user is requested to select a time interval to be used for analysis.

Once this has been completed, the program displays a list of functions that can be performed on the data. These functions are averaging, autocorrelation, filtering routines, store data, plot data or exit the program.

5.4.1.9.2 Averaging

The averaging routine is not currently implemented. Several methods of averaging were tried, but all have failed to produce an acceptable average. Averaging methods attempted are described in detail in the section 6.1.1.

5.4.1.9.3 Autocorrelation

The autocorrelation routine computes the autocorrelation function of the selected time interval. The maximum number of points that can be used in this function is 10,000. Attempts at correlating more the 10,000 points will result in correlation for only the first 10,000 points. Also if the time interval is near the end of the data run, then less than the desired number of correlation points may be returned. This is due to the fact that in order to autocorrelate 5,000 points, 10,000 points are needed.

5.4.1.9.4 Filtering

The filter option will list a submenu that allows the user to choose the type of filter he would like to use on the data. If the lowpass or highpass options are picked, the user will be prompted for the order and the cutoff frequency in Hz of the filter. The bandpass and band reject filters will prompt for the order, the bandwidth in Hz, and the center frequency in Hz. The store and plot functions in this submenu will store or plot the current filtered data.

5.4.1.9.5 Storage and Plotting

The store and plot functions will check to see which options have been previously executed on the data and give a corresponding menu. The user will then make his choice and the correct output will be either stored to a file or plotted.

5.4.2 Data and file structure documentation

The data and file structure documentation adheres to the structured approach. The file structure used in this program, reflects the HP-UX operating system origin, a UNIX derivative. The HP-UX file system is a hierarchical, "treelike" structure which allows users to organize files in a convenient and logical fashion.

5.4.2.1 Data description

The documentation of the data consist of a description of the input and output variables required by the program module. The assumption is that the digitized acoustic, position,and rotor speed data are available in suitable files for program input. The data streams have been synchronized in time and there is a corresponding position and rotor speed data point for each acoustic data point.

The file structure which is attached to the operational modules consists of the 'header' and 'data' files. The 'header' files contain the descriptive information about the data in 'data' files. Each module requires at least one 'header' and one 'data' file for input. Each module will also output a 'header' and one or more 'data' files.

The description of the data variables will be made only for the modules which require external data input to run. The inter module data description will not be made because it does not provide the user with useful information.

5.4.2.1.1 Module 'INPUT'

The 'INPUT' module is the first module which is used to initialize the program and input the global type variables. The file naming convention used for identifying the program data structure are string variables 'JOBNAME' and 'RUN'. Other variables are numeric and are used in computations. The files which comprise the input data set are stored in the directory /users/de-doppler/JOBAME/RUN/ input.

5.4.2.1.1.1 File 'hinput'

This is a header file containing the variables used in the computation by many of the functional modules.

VARIABLE	TYPE	FORMAT	DESCRIPTION
JOBNAME	string	A14	JOBNAME is used to define root directory under which the data is stored.
RUN	string	A14	RUN is used to track a particular data set which is being restored. The run is coded to show microphone location. Coding: 1-5 run number; 6-7 microphone position: 04 side line south; 05 Center line 06 side line north; 8-14 description
RPM	real	F12.2	RPM is the reference main rotor speed.
TEMP	real	F10.1	TEMP is the air temperature in degrees C.
RO	real	F10.1	RO, in m, is the reference distance at the origin.
MIC	real	F10.1	MIC is an array describing the microphonelocations relative to the coordinate axis. The three coordinates X,Y,Z, in m, are required for each microphone location. Default is three standard microphone locations used in rotorcraft certification testing.
NOPS	int	I7	NOPS is the number of data points to be used in the computation.
DELT	real	F12.8	DELT is the interval between the measured data samples in microseconds.
DELT2	real	F12.8	DELT2 is the interval between the restored data points in microseconds.
STTIME	int	I6	STTIME is the array containing the starting time of the data stream. It is in the form hours: min: sec: milliseconds: microseconds.

5.4.2.1.1.2 File 'acoustic'

This file contains the sound pressure data to be used in the restoration process.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is a array of sound pressure in Pa.

5.4.2.1.1.3 File 'x'

This file contains the x coordinate of the position data.

VARIABLE	TYPE	FORMAT	DESCRIPTION
X	real	F12.6	X is a array of x coordinate in m.

5.4.2.1.1.4 File 'y'

This file contains the y coordinate of the position data.

VARIABLE	TYPE	FORMAT	DESCRIPTION
Y	real	F12.6	Y is a array of y coordinate in m.

5.4.2.1.1.5 File 'z'

This file contains the z coordinates of the position data.

VARIABLE	TYPE	FORMAT	DESCRIPTION
Z	real	F12.6	Z is an array of z coordinates in m.

5.4.2.1.1.6 File 'rpm'

This file contains the main rotor speed data.

VARIABLE	TYPE	FORMAT	DESCRIPTION
RPM	real	F12.6	RPM is a array of the main rotor speed data in RPS.

5.4.2.1.2 Module 'TIMEDOMAIN'

The 'TIMEDOMAIN' module is used to perform time domain operations. The input data structure for this module is not useful to the user, hence only the output file structure will be discussed. Each time domain operation produces two files, a header file and a data file. The 'header' file contains the information necessary for

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further use of the data contained in the data file. The files which comprise the output are stored in the directory /users/de-doppler/JOBNAME/RUN/postprocess/.

5.4.2.1.2.1 File 'htime'

This is a header file which contains the information about the output from the time interval selection operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
NUMBER	int	I7	NUMBER is the number of the data points selected by the user.
DT	real	F12.8	DT is the time interval between the data points in microseconds.

5.4.2.1.2.2 File 'time'

This is a data file which contains the time history of the sound pressure in Pa.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is the array of sound pressure data selected by the user.

5.4.2.1.2.3 File 'haverage'

This is a header file which contains the information about the output from the ensemble averaging operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
NUMBER	int	I7	NUMBER is the number of the data points in the ensemble averaged array.
DT	real	F12.8	DT is the time interval between the data points in microseconds.
ENSEMBLE	int	I7	ENSEMBLE is the number of the ensemble averages.

5.4.2.1.2.4 File 'average'

This is a data file which contains tha averaged time history of sound pressure in Pa.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is the array of sound pressure data selected by the user.

5.4.2.1.2.5 File 'hauto'

This is a header file which contains the information about the output from the autocorrelation operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
NUMBER	int	I7	NUMBER is the number of the data points in the autocorrelation array.
DT	real	F12.8	DT is the time interval between the data points in microseconds.

5.4.2.1.2.6 File 'auto'

This is a data file which contains the result of the autocorrelation operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is the array of autocorrelation data.

5.4.2.1.2.7 File 'hfilter'

This is a header file which contains the information about the output from the filtering operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
NUMBER	int	I7	NUMBER is the number of the data points in the filtered array.
DT	real	F12.8	DT is the time interval between the data points in microseconds.
FTYPE	string	A20	FTYPE string variable containing the type of the filter used.
FLIM	real	2F12.6	FLIM is an array containing the lower, upper cutoff frequencies, and/or band bass limits of the filter.

5.4.2.1.2.8 File 'filter'

This is a data file which contains the result of the filter operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is the array of sound pressure in Pa. after filtering.

5.4.2.1.3 Module 'PSD'

The 'PSD' module is used to perform FFT operations on the time domain data and compute the power spectral density, PSD. The input data structure for this module is the output from the 'TIMEDOMAIN' module. The header file contains the data which describes the contents of the data file. The files which comprise the output of the 'PSD' module are stored in the directory /users/de-doppler/JOBNAME/RUN/postprocess/.

5.4.2.1.3.1 File 'hPSD'

This is a header file which contains the information about the output from the PSD operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
NUMBER	int	I7	NUMBER is the number of the data points in the PSD array.
DF	real	F12.8	DF is the frequency interval between the data points in Hz.
ENSEMBLE	int	I7	ENSEMBLE number of ensembles.
ST	real	F12.6	ST is the array containing the stating time of the array.
WINDOW	string	A20	WINDOW is a string describing the type window used in FFT process.

5.4.2.1.3.2 File 'psd'

This is a data file which contains the result of the PSD operation.

VARIABLE	TYPE	FORMAT	DESCRIPTION
DATA	real	F12.6	DATA is the array of PSD values in Pa^2/Hz.

5.4.2.2 Data Structure Description

The documentation of the data structure is a description of the input and out files required by the program modules and their location in the hierarchical file system. As was previously mentioned, the software was developed under the HP-UX operating system. At the head of the file system is the so called 'root' directory from which the file system grows. The root directory consists of a number of directories which in turn may contain data files and/or other directories.

Unfortunately FORTRAN 77 language does not recognize the difference between the directory files and data files: therefore the programs which are coded in FORTRAN do not use the hierarchical file structure feature available under HP-UX. The absence of the ability to use relative addressing or change directory commands without resorting to the system calls or C language routines, results in unnecessarily long file names.

5.4.2.2.1 Module 'INPUT'

The input module has no input files, but only the output file 'header'. This file contains global type of information which is described in 5.4.2.1.1. This file is located in the directory/ users/de-doppler/JOBNAME/input/RUN.

5.4.2.2.1 Module 'TIMEDOMAIN'

The 'TIMEDOMAIN' module has input files, which consist of the 'header' and 'data' files. These files are located in the directory /users/de-doppler/JOBNAME/restored/RUN. The output files are stored in the directory/users/de-doppler/JOBNAME /postprocess/RUN. The input file structure is not important for the user because he has no control over the content of the files. The output files are described in detail in 5.4.2.1.2.

5.4.2.2.2 Module 'PSD'

The 'PSD' module has input files, which consist of 'header' and 'data' file. These files are located in the directory/users/de-doppler/JOBNAME/postprocess/RUN. The output of this module stores files in the directory/users/de-doppler/JOBNAME /postprocess/RUN. The input file structure is not important for the user because he has no control over the content of the files. The output files are described in detail in 5.4.2.1.3.

6.0 RESULTS

6.1 General Discussion of Results

The signal restoration has been achieved with satisfactory results in the frequency domain and partially successful in the time domain. The best way to demonstrate that the programmed restoration algorithm works is to process a long data sample where the Doppler shift rate of change changes in sign, then perform FFT and ensemble average the whole data run. If the methodology is successful the resultant PSD spectrum should show no evidence of smear due to frequency modulation. The amplitude should not show the effects of the amplitude modulation. The result of such a test can be seen by comparing the de-Dopplerized spectrum in Fig. 8 with the non de-Dopplerized spectrum in Fig. 9. The improvement in the signal to noise ratio and relative absence of the smear in the frequency domain is self evident. In Fig. 10 one can clearly see the harmonics and the side bands of the main and the tail rotors. The interaction tones which are very difficult to observe can be clearly identified in Fig. 10. The time domain signal has been properly restored for the modulation effects, and is shown in Fig. 11. The time domain averaging was not successful (see explanations in 6.1).

6.2 Time Domain Results

The failure to achieve an acceptable synchronous average in the time domain is traced to the inadequate main rotor speed resolution. The main rotor speed signal was updated each 0.5 seconds and interpolation was used to create the intermediate data points. This approach is clearly inadequate judging by the results.

Therefore to be able to perform averaging in the time domain, an appropriate method must be implemented for synchronization between the sampled data and the rotor speed. Several attempts where made to achieve this synchronization but the results were disappointing. It is evident that a rotor speed signal with a high sampling rate is required to achieve the necessary degree of synchronization.

6.2.1 Time Domain Averaging Methods Attempted and Results

Attempts were made to utilize the same data set for frequency and time domain averaging. The first attempt at averaging involved obtaining the main rotor period from the output of the FFT routine. The object of this attempt was to perform a so called synchronous average. Then block averaging was performed, where the block length was equal to one period of the main rotor.

This method was successful in obtaining an average over one quarter of the period (one blade pass), but when it was extended to calculate the average of a full period, the average approached zero. The problem was traced to the rotor speed variation over one period of rotation and the consequent variation in the period of the rotation. This seemingly small speed variation is sufficient to make the averaging process non-synchronous.

Block averaging was attempted using a random block length. The random block length produced an average near zero. Hence the method was discarded.

The next attempt was to use a block length which is based on the emission angle measured in radians. This method is sometimes used in directivity studies where an average is formed over a given emission angle. This method produced very small peaks in the averaged data, but they were extremely close to zero and almost negligible. The possible cause of failure was once again due to the non-integral number of the main rotor periods within the sampling period.

6.3 Frequency Domain Results

The frequency domain results have met all of the objectives established for this signal restoration program. The restoration of a complete flyover shows that the restoration procedure has removed the major portion of the signal smear due to the source motion effects. The comparison of the de-Dopplerized, Fig. 8, and non de-Dopplerized, Fig. 9, spectra of the entire flight event supports the above conclusion. The residual lack of sharpness in the spectrum as compared to typical wind tunnel test spectra is due to the secondary modulation effects such as wind, turbulence, and refraction effects. These effects were considered to be secondary in nature and the results tend to sustain this conclusion.

The spectrum shows that the flyover spectrum is dominated by the main rotor, its harmonics and side bands, these are shown in Fig. 10. It is important to note that at higher frequencies, side bands sometimes exceed the level of the harmonics. The tail rotor spectrum shows, in addition to the harmonics, the presence of strong side bands below the blade passage frequency. The sidebands are about equal in level to that of the blade passage frequency tone.

In addition to the spectral structure composed of the harmonics and side bands of the main and the tail rotors, there is a strong evidence of the interaction tones between the main and the tail rotor in the region between 300 and 400 Hz (see Fig. 10). Without the signal restoration these effects could not have been extracted from the flyover data by conventional signal analysis methods. The

usual assumption of "locally stationary" which was used for narrowband spectral analysis has sufficient smear and lack of statistical confidence so that the interaction tones cannot be reliably identified.

The restoration method has shown that the tonal nature of the rotorcraft noise in flight is similar to the one observed in the wind tunnel. Hence flight testing now can be used for source identification by using this program. In order to obtain a more detailed picture of the rotorcraft noise signature, such as directivity, the signal restoration should be coupled with a linear array methodology for directivity measurement. The linear array would also make available the ensemble averaging process for increasing the data confidence level. The time domain restoration should be used to preprocess the data acquired by the linear microphone array. The coupling of the two techniques will lead to more detailed rotorcraft acoustic data with higher confidence level.

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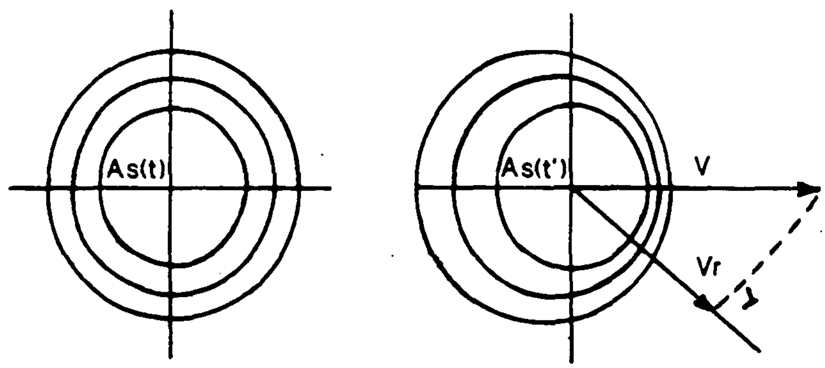


$\lambda = cT = c/f$

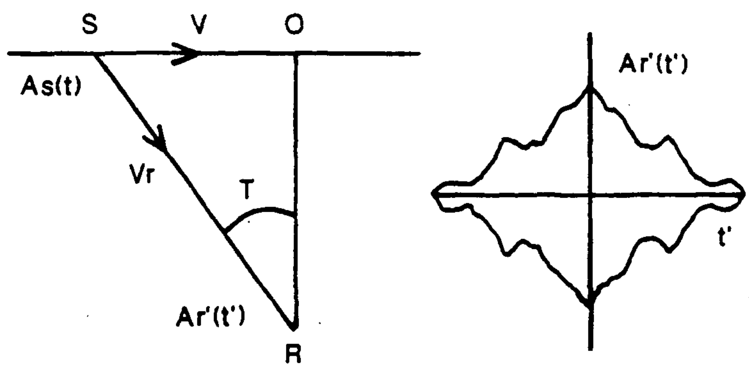
$\lambda' = \; - Vr\, T$

$T = (1 - Mr)\, T$

$Mr = Vr/c$

STATIONARY SOURCE   MOVING SOURCE

FIGURE 1. EFFECT OF MOTION ON THE SOURCE SIGNAL DOPPLER SHIFT.



SOURCE S MOVING AT SPEED V
RADIATING SIGNAL As(t)

Ar'(t') IS THE SIGNAL AT R
FROM MOVING SOURCE AT S

FIGURE 2. AMPLITUDE MODULATION DUE TO SOURCE MOTION.

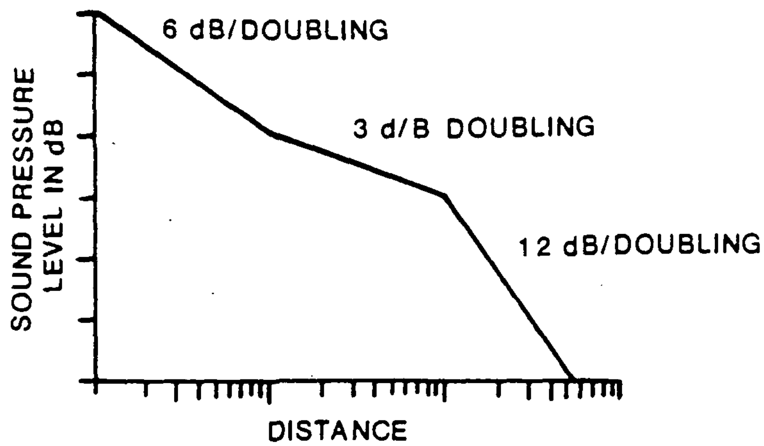


FIGURE 3. CHARACTERISTICS OF GROUND WAVE PROPAGATION.

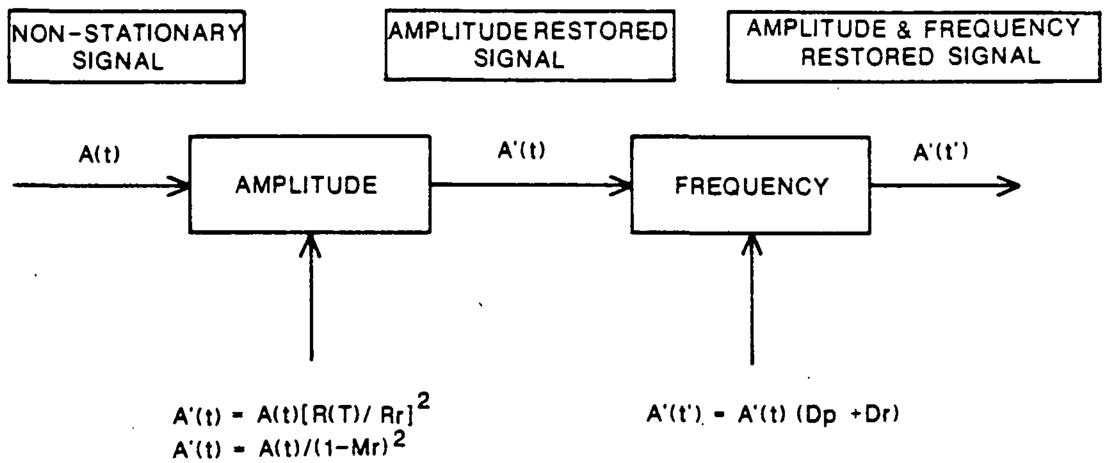


FIGURE 4. SIGNAL RESTORATION PROCESS.

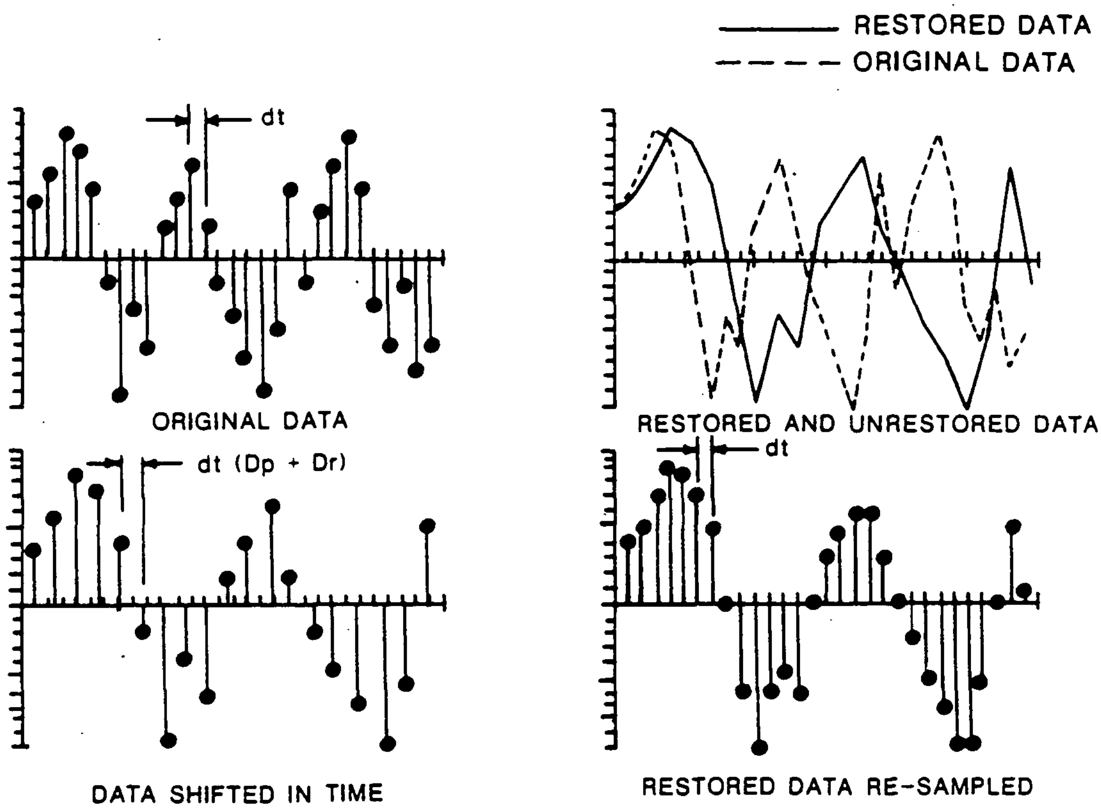


FIGURE 5. RESTORATION OF FREQUENCY MODULATION.

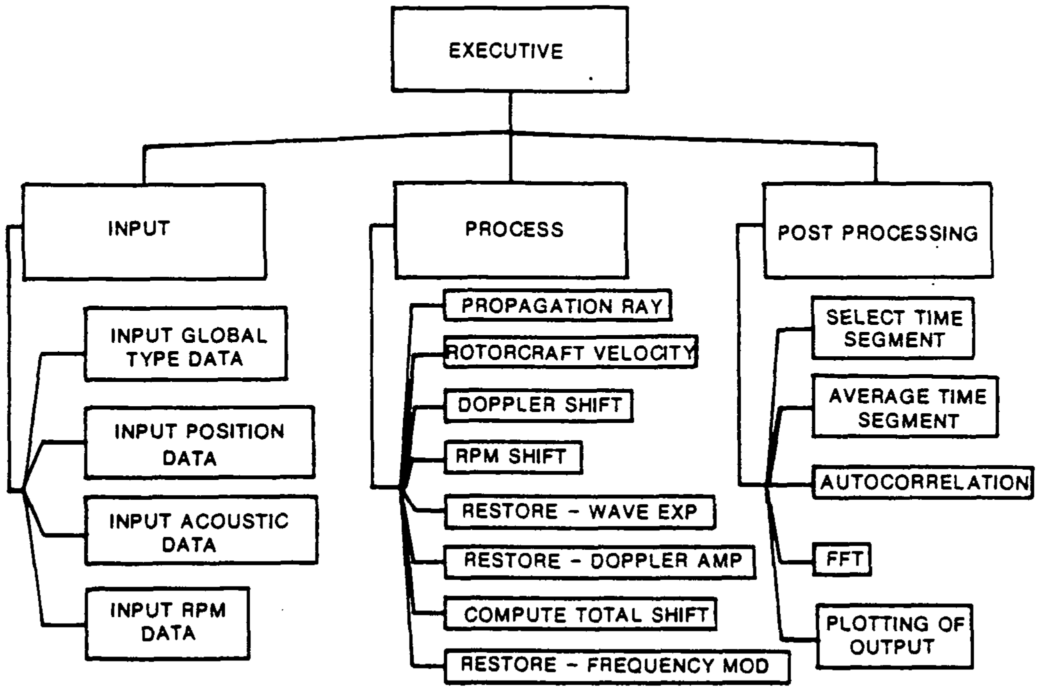


FIGURE 6. STRUCTURE OF SIGNAL RESTORATION SOFTWARE.

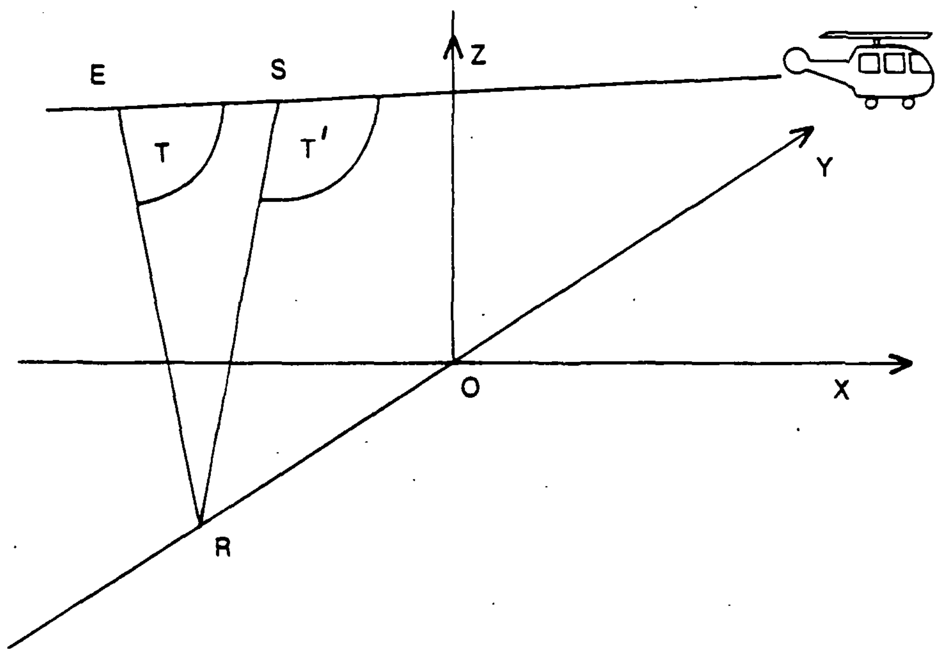


FIGURE 7. FLIGHT PATH GEOMETRY FOR RESTORATION PROCEDURE.

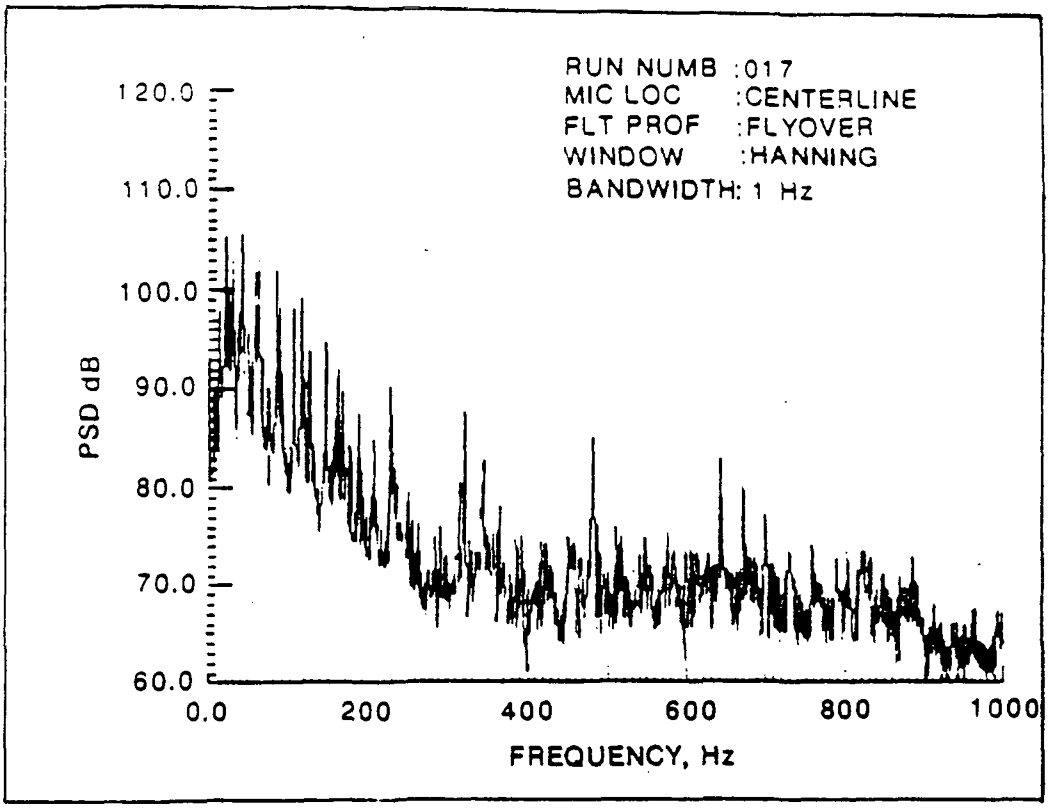


FIGURE 8. DE-DOPPLERIZED SPECTRUM.

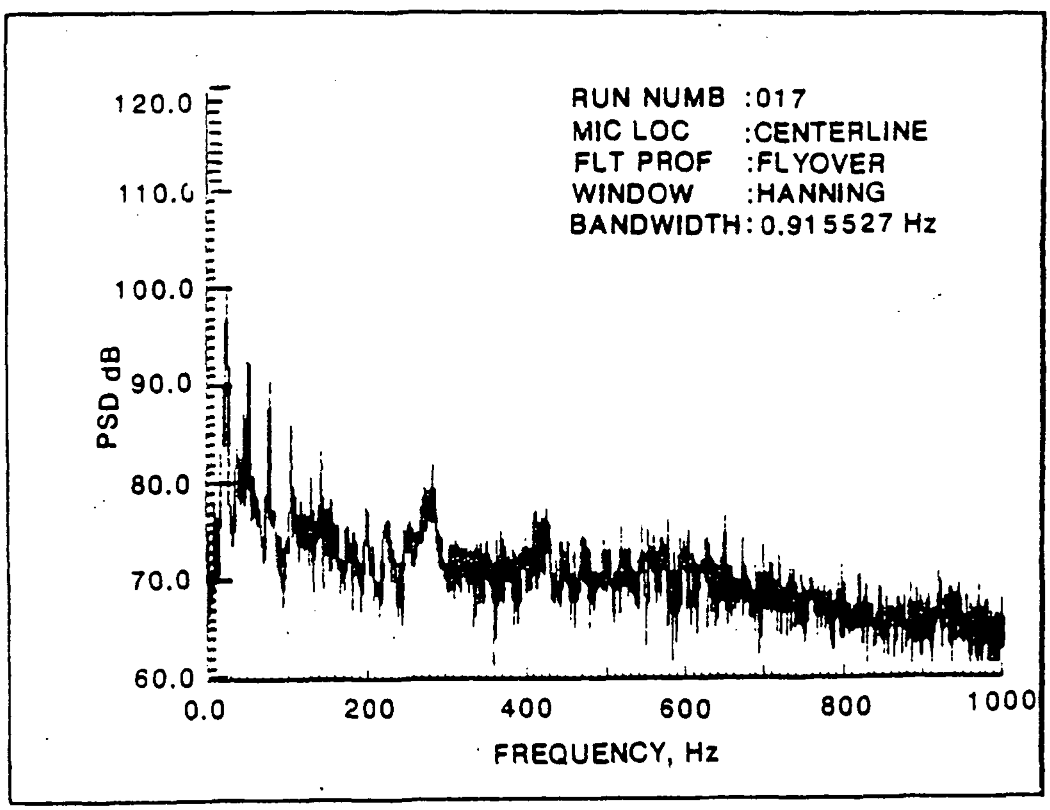


FIGURE 9. NON DE-DOPPLERIZED SPECTRUM.

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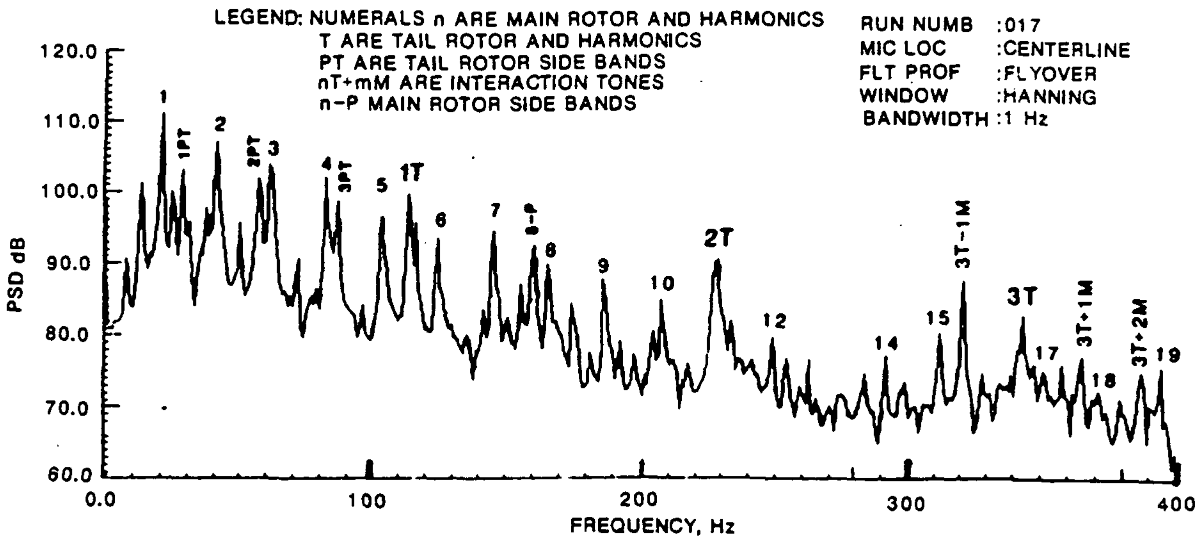

FIGURE 10. RESTORED FLYOVER SPECTRUM.

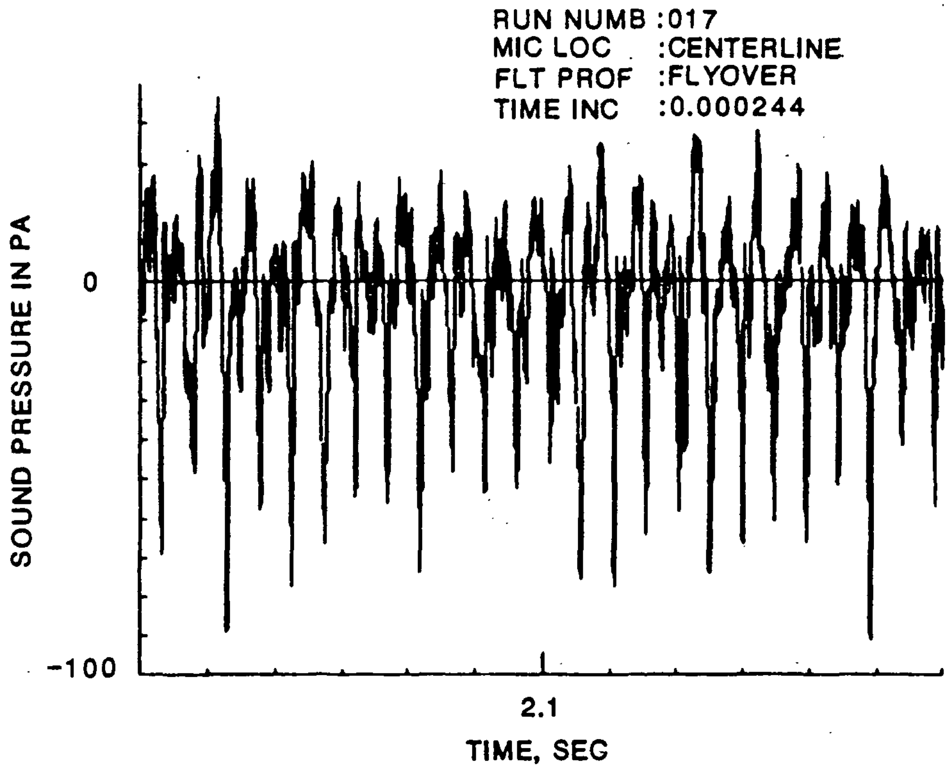

FIGURE 11. RESTORED FLYOVER TIME HISTORY.